



AMENDMENTS TO THE CLAIMS

7217/70982

This listing of claims will replace all prior versions and listings of claims in the application;

Claim 1. (Currently Amended) An audio signal processing method comprising the steps of:

supplying an audio signal to each of a plurality of digital filters;

respectively supplying outputs from the plurality of digital filters to ~~each~~ of a plurality of speakers forming arranged in a speaker array to form a sound field;

setting a predetermined delay time ~~to be given~~ in each of the plurality of digital filters so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of the plurality of speakers will coincide with each other; and

adjusting ~~the~~ an amplitude characteristic of the plurality of digital filters so that a low-pass filter characteristic will be given to ~~the~~ a synthesis response of the audio signal at a second point in the sound field.

Claim 2. (Currently Amended) The audio signal processing method according to claim 1, wherein a sound wave from the speaker array is caused to reach at least one of the first and

7217/70982
second point points after it is reflected by a wall surface.

Claim 3. (Currently Amended) The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound ~~filter~~ field, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the ~~latter~~ plurality of digital filters.

Claim 4. (Original) The audio signal processing method according to claim 1, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters.

Claim 5. (Currently Amended) The audio signal processing method according to claim 1, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time is ~~over-sampled~~ for a shorter period than the a

7217/70982
sampling period to provide a sample train and, wherein the sample train is down-sampled to provide pulse-waveform data of the sampling period; and

the factor data is set for a part to be delayed by the plurality of digital filters based on the basis of the pulse-waveform data.

Claim 6. (Original) The audio signal processing method according to claim 5, wherein the audio signal is delayed by a part of the predetermined delay time, which is a multiple of the sampling period, by digital delay circuits which operate for the sampling period, while it is being delayed by the remainder of the predetermined delay time, which includes the decimal part by the digital filters.

Claim 7. (Currently Amended) The audio signal processing method according to claim 5, wherein:

the an over-sampling period of the over-sampling operation is $1/N$ (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

7217/70982
Claim 8. (Currently Amended) The audio signal processing method according to claim 7, wherein:

the pulse-waveform data to be delayed by a delay time which is m/N ($m = 1$ to $N - 1$) of the sampling period is pre-stored in a data base; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

Claim 9. (Currently Amended) The audio signal processing method according to claim 5, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

Claim 10. (Currently Amended) An audio signal processor comprising a plurality of digital filters each supplied with an audio signal, wherein

each of the plurality of digital filters supplies an output signal to each of a plurality of speakers forming arranged in a speaker array to form a sound field;

each of the plurality of digital filters has a predetermined delay time so that transmission delay times with which the audio signal arrives at a first point in the sound field via each of the plurality of digital filters and each of

7217/70982
the plurality of speakers will coincide with each other; and

each of the plurality of digital filters has an amplitude characteristic so that a low-pass filter characteristic will be given to the a synthesis response of the audio signal at a second point in the sound field.

Claim 11. (Currently Amended) The audio signal processor according to claim 10, wherein a sound wave from the speaker array is caused to reach at least one of the first and second point points after it is reflected by a wall surface.

Claim 12. (Currently Amended) The audio signal processor according to claim 10, wherein when forming the first and second points in the sound filter, a filter factor of each of the plurality of digital filters is determined by calculation and set for each of the latter plurality of digital filters.

Claim 13. (Original) The audio signal processor according to claim 10, wherein when forming the first and second points in the sound field, a filter factor of each of the plurality of digital filters is read from a data base and set for each of the plurality of digital filters.

Claim 14. (Currently Amended) The audio signal processor according to claim 10, wherein:

7217/70982
the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal,

there is further provided a calculation circuit to calculate pulse-waveform data of the sampling period by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform provided by the calculation circuit is set as a filter factor of each of the plurality of digital filters.

Claim 15. (Currently Amended) The audio signal processor according to claim 14, wherein:

~~the an~~ over-sampling period of the over-sampling operation in the calculation circuit is $1/N$ (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

Claim 16. The audio signal processor according to claim 14, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set synthetic-waveform data as a filter factor of each of the plurality of digital filters.

Claim 17. (Currently Amended) The audio signal processor according to claim 10, wherein:

the predetermined delay time set for at least one of the plurality of digital filters is divided into an integer part and decimal part in units of a sampling period of the audio signal;

there is further provided a storing means for storing pulse-waveform data of the sampling period provided by over-sampling an impulse response including a delay time represented by at least the decimal part of the predetermined delay time for a shorter period than the sampling period to provide a sample train, and down-sampling the sample train; and

the pulse-waveform data stored in the storing means is taken out and set as a filter factor of each of the plurality of digital filters.

Claim 18. (Currently Amended) The audio signal processor according to claim 17, wherein:

the an over-sampling period of the over-sampling ~~operation~~ is $1/N$ (N is an integer larger than or equal to 2) of the sampling period of the digital signal; and

when the delay time represented by the decimal part is nearly an integral multiple (m) of the over-sampling period, m/N is adopted as the decimal part.

Claim 19. (Currently Amended) The audio signal processor according to claim 17, wherein:

a plurality of the pulse-waveform data corresponding to the decimal part is pre-stored in the storing means; and

pulse-waveform data approximate to the decimal part is taken out of the stored pulse-waveform data and set as a filter factor of each of the plurality of digital filters.

Claim 20. (Currently Amended) The audio signal processor according to claim 17, wherein a transfer characteristic providing a predetermined acoustic effect is convoluted in the pulse-waveform data to set the pulse-waveform data as a filter factor of each of the plurality of digital filters.